Testing the installed sound system: Testing a system helps identify hard-to-find yet easy-to-correct installation errors that ruin the sound of your system.

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In all areas of life, we rely on goods and services to make our lives easier and more productive. And with our investment, we have the right to expect a level of performance consistent with what was promised by the manufacturer or designer. Final testing and calibration plays an integral part in assuring that the product or service has been manufactured or installed in a manner that assures that the full usefulness of the device is realized. Testing the installed sound system in a methodical and repeatable manner benefits the end-user because the system performs better; testing benefits the designer and installer by revealing shortcomings that lead to better future designs.

What to evaluate A common misconception is that the installer should test the system after the system is in place. In fact, you should test a system during the installation process. That way problems can be detected and corrected as they occur. We're assuming we are working with a good design and good components. From here the focus shifts to assuring they work together properly.

During the installation and commissioning process, the installer should evaluate a number of areas for proper operation. They include:

Safe rigging practices.

Safe grounding practices.

Hums resulting from ground currents.

Proper system polarity.

Susceptibility to radio-frequency interference.

Susceptibility to electromagnetic interference.

Proper system gain structure.

Consistency and accuracy of coverage.

Fidelity of the direct sound field LD.

Adequate acoustic gain.

Of all of these, the most important are rigging and safety grounding. Even systems that soundgood can kill or maim if not properly affixed and properly grounded. If in doubt about the rigging, defer to a specialist in that field. Be certain that all loudspeakers have safety cables affixed and securely fastened. Safe grounding practice means each component is earth grounded through its AC cord, rack rails or both. Improperly grounded systems can present a shock hazard to the end user, and the results are often lethal. The challenge--to accomplish safety grounding without introducing audible hum into the system--is certainly possible, although it may require minor modifications to some system components. The final words in these two critical areas: If in doubt, consult an expert.

Ground current Ground current problems most often result from current flowing around multiple ground paths between electronic devices. Whenever wire is looped, current flow will be induced by fields that radiate from electrical devices or conductors in close proximity. Also, since different parts of a sound system will have slight variations in the potential of their earth grounds, current will flow between these ground points. Just like water, electricity will always try to level itself. Water covers 70% of our planet, yet there is only one reference that we call sea level. Any body of water with a connection to the sea will seek that level. That's why rivers flow and lakes don't.

Sound-system grounding works much the same way, with an earth ground that is the zero potential (or sea level) point of the system. All other ground points higher than this potential will seek this one when they are connected together. Care should be taken when using an electrical circuit that is not dedicated for the audio system with its own technical ground connected at the building service entrance. Even with a good technical ground, there are often three ground paths between individual electronic components, which makes the flow of ground current inevitable. Sound system components are connected together through the AC supply for the system, the rack rails and the shields in the interconnect cables. Ground currents flowing through these multiple paths often find their way into the signal path, resulting in audible hum or buzz at the output of the system. The problem is not that the current is flowing, but in where the current is flowing. When it finds its way onto circuit boards, things are going to buzz.

You can take two approaches to correct this problem. One is to try to eliminate all but one of the ground paths. This approach can be frustrating and often results in unsafe practices, such as lifting the third prong of the AC electrical cord. Other methods include isolating equipment from rack rails with insulating washers and lifting shields at one end on interconnect cables. All of these are stop-gap measures and never really address the heart of the problem: Ground current hums are caused by current flowing around ground paths and into the internal circuitry of mixers and amps; how does it wind up there?

Unfortunately, many manufacturers provide a path directly onto the circuit boards by connecting their internal circuit grounds to pin one on balanced input and output connectors. This provides a direct route for all kinds of nasties to invade the most sensitive circuits of a device. Imagine routing the downspouts from the gutters on your house into your basement; connecting cable shields to signal grounds is the electrical equivalent of this plumbing disaster. One way to break the flow--lifting interconnect cable shields at one end--can also cause problems because the lifted shield becomes an antenna that may be connected directly to a circuit board via pin one on an input or output connector at the un-lifted end of the cable.

Because the cable shield is essentially the "metal chassis" that surrounds the signal conductors, the best solution is to connect all shields to AC or chassis ground at both ends of the interconnect cables, which means connecting the shields directly to the metal box that houses the equipment. Some manufacturers do this for you, but as of the time of this writing, most do not. As a result, an important part of the interconnect procedure is to identify and correct pin-one problems. The pin-one problem has been well-documented by Neil Muncy and other grounding evangelists, to whom the audio industry owes a debt of gratitude.

A simple tester works well to identify pin-one problems. (See Figure 1.) Called a hummer by its inventor, John Windt, this inexpensive tester deliberately injects about 100 mA of 60 Hz AC current into pin-one of each system component. Properly grounded inputs and outputs will not conduct this ground current onto circuit boards. Improperly grounded components will hum at the presence of this electrical invader. A current of 100 mA is chosen because it is a typical value of ground current that will flow in a large sound system. To perform this test, start at the power amplifier and work your way back to the system input as you interconnect components. My hummer has an XLR-type male connector that can be plugged into the input jack of each component. If a component hums when tested in this manner, reroute shields on interconnect cables directly to the metal chassis rather than pin-one on both the input and output jacks. (See Figure 2.)

Seem like a lot of trouble? Finding and correcting a ground loop in a large system is much more trouble and will take much longer to correct. The installation phase of a project is the time to do it. Not only will you find the problem or potential problem, you will educate yourself as to which manufacturers properly ground their inputs and outputs, saving you time and money in the future.

Polarity Maintaining proper polarity is important in every part of the system, especially in loudspeaker arrays because of the close spacing (and therefore significant acoustic coupling) of loudspeaker components.

Polarity problems are most easily understood if you consider the low-frequency loudspeakers of a sound system, because we can sometimes actually see them move. Of course the high-frequency devices are moving too, just faster and with smaller excursions (and invisible to the naked eye). When a positive-going signal is applied to the positive loudspeaker terminal, the loudspeaker should move outward in a properly polarized sound system. From a system perspective, an over-pressure at the microphone diaphragm should produce an over-pressure at the listener; as the microphone diaphragm moves in, the loudspeaker diaphragm should move out.

Suppose you had two loudspeakers in close proximity. One moves outward in response to a signal; the other moves inward. The net effect is no sound or very little sound because each movement is telling the air molecules to do the opposite thing. The actual result is frequency dependent and usually manifests itself as weak bass because of longer wavelengths and increased acoustic coupling at low frequencies. (Remember, the lower the frequency, the longer the wavelength and the greater the acoustic coupling between closely spaced transducers.)

For subwoofers, the cancellation can be nearly complete. In mid- and high-frequency loudspeakers, polarity problems often show up as a dead spot at a location equidistant between the two devices that are out of polarity, or an acoustic image that appears to be coming from somewhere other than where the loudspeakers are located. Mispolarized loudspeakers often create floor or ceiling reflections that are stronger than the direct field at the listener, so accurate imaging goes out the door.

Polarity can be tested several ways, ranging from quick and dirty to quite sophisticated. Let's start with a quick and dirty method that works well for woofers. Simply take a 9 V battery and touch it to the loudspeaker terminals. The loudspeaker should move outward when the positive battery terminal is connected to the positive loudspeaker terminal. Test the cable in the same manner, with one end connected to the loudspeaker.

High-frequency devices aren't quite so easy. The best method here is to use a set of polarity testers, which consists of a send and receive unit. The send unit outputs a pulse, and the receive unit--you guessed it--receives it. The outgoing pulse is a positive-going spike, and the receive unit, which has a built-in microphone, shows whether the received pulse is positive (proper polarity) or negative (reversed polarity). These testers are commercially available from a number of sources.

Work through the system in the following manner, beginning at the system output--the loudspeaker--and working toward the input--the microphone. An ideal time is right after the hummer test is conducted on each component. First, place the receive unit in front of the loudspeaker (about a foot or 250 mm is a usually a good distance). Use the send unit to drive the loudspeaker directly (some have a "loudspeaker out" jack especially for this purpose). Once the proper polarity of all loudspeaker devices has been established, you can then work backward as you interconnect the electronic devices ahead of the loudspeaker. This includes power amplifiers, crossovers, equalizers, mixers, microphones, etc. Be sure to check all interconnecting cables. I have found many reverse-polarity cables right off the shelf. Be sure to check each signal-processing device in both the standard and bypass modes, as some devices invert polarity when they are switched in or out.

Also, remember we are looking for devices that invert polarity between their input and output. At least for this purpose, it doesn't matter if a device is pin-two high or pin-three high, as long as it doesn't invert the polarity of the signal. This is true for all devices except power amplifiers, as a pin-three high-power amp will invert the polarity of the signal sent to the loudspeaker. I wonder how many mysterious sound system problems have been solved by overhauling the system when the main problems were actually simple polarity reversals.

As with pin-one problems, the time to test is during installation; otherwise when mysterious problems do arise (as they always do), the first thing that must be done is to rule out polarity reversals. For instance, if the sound system has poor bass response caused by polarity inversion between two subs, you may incorrectly determine that new subs are required when all you really need to do is reverse two wires.

Microphones are another major offender concerning polarity because the output jacks are often wired by hand on an assembly line. Again, you can find the "Monday morning microphone" with a polarity tester or by placing it in close proximity to a microphone you know is good and speaking into both. Identically polarized microphones will sum when set at equal levels; reverse-polarity microphones will cancel. Keep a "golden microphone and cable" of known polarity on hand as a standard.

Radio-frequency interference Radio-frequency interference (RFI) is another problem that plagues sound systems. Fortunately, most RFI problems will be corrected by fixing the pin-one problems already described. Good grounding practice means that extraneous noise induced into a sound system has a low-impedance path directly to ground through cable shields and never gets to sensitive circuit components, much like a good drainage system around your house. This problem goes beyond having country music hits pouring out of a system without a tuner installed. Sound systems that have RFI problems will pop and click as electrical contacts close throughout the building. The small spark caused by switching a light on or off generates a broadband burst of energy whose bandpass extends into the radio frequency portion of the spectrum.

A good way to test for this is to generate a small electrical spark near the equipment rack and listen for clicks at the output of the system. A gas grill ignition or small electric fence-charger can be used for this purpose (be VERY careful). Keep a distance of 5 to 10 feet (1.5 to 3 m) between the spark source and any system component, and don't use it around DSP-based equipment. Use a headphone amplifier to isolate the problem to a specific component or microphone line. RFI filters are sometimes useful at system interconnect points. (See Figure 3.)

Electromagnetic interference (EMI) Even with all pin-one problems eliminated, hum can still get into a sound system by induction. A common culprit is a signal processor mounted in close proximity to a power amplifier. The electromagnetic field from the amplifier induces hum components into the sensitive circuitry of the processor. The best fix for this problem (rack manufacturers will love this) is to isolate the two devices physically. It is also a good idea to buy equipment that carries the CE label, which means that it has been tested and approved for sale in Europe. Our European friends have strict guidelines on the permissible fields that can radiate from an electrical device. Devices that meet these guidelines will be less likely to have EMI problems.

One way to locate a hum-sensitive component is to use a degaussing coil or a tape-head demagnetizer, both of which are available from the larger electronic supply houses. Move the coil in close proximity to each system component while monitoring the output of the system. This procedure has about the same affect on floppy disks as attaching them to the refrigerator with a magnet, so keep it away from computers and other devices that employ magnetic storage, unless you just enjoy having a system crash and burn when you turn it on.

Coverage problems In any installed sound-reinforcement system, there are inevitably good seats and bad seats in the audience. For speech reproduction, good seats result from a design that provides the following:

Time-coherent direct sound field.

Good signal-to-noise ratio.

Good direct-to-reverberant ratio.

Absence of strong reflections less than 5 ms or greater than 40 ms after the arrival of the direct sound field.

For music reproduction, no one can really decide what sounds right, which is why we have marketing departments to tell us what we should like. Yes, "warm bass sound" is just another way to say "distortion."

When a listener seat has poor speech intelligibility, one or more of these criteria has been violated. Our ears are excellent at telling us that a problem exists, but we sometimes need supplemental information as to why a problem exists. Welcome to the world of audio and acoustic measurements.

Most sound-system measurements fall into two categories: electrical measurements and acoustical measurements. Signals in the electrical domain are referred to as audio signals. Signals that propagate by vibrating a material such as air, water or steel are called acoustic signals. Audio signals are measured by direct connection to the analyzer. Acoustic signals must be acquired via an appropriate transducer: the golden measurement microphone. A good microphone is required because its response will be superimposed on the data you collect.

Manufacturers offer many good audio and acoustic measurement systems. Some are so simple that a novice can use them; others require special training and expertise. Let's look at the pros and cons of a few of the major types.

Single-port measurement systems gather data only, with no knowledge of the input signal to the system. You have two of these attached to your head. Sound-level meters and real-time analyzers also fall into this category. They usually run off batteries (except for your ears, which are food-powered) and can be useful for quick checks and verifying coverage over large areas.

An instrument that measures sound level only is called a sound-level meter. Such meters use a pressure-calibrated microphone to measure fluctuations in sound pressure, then display the information as a sound level in decibels on an analog or digital display. Sound-level meters that display the spectral response of the system and the room are called real-time analyzers.

The most popular type has filters spaced at 1/3-octave intervals. They are easy to use but are unable to ignore ambient noise, reflections and reverberation when displaying the response of the system. Also, the analyzer displays the spectral content of the program material, so you are not really measuring the sound system, but the combination of the room, sound system and whatever is being fed into it.

To provide a more consistent program source, noise can be input into the system. Suitable noise, called pink noise (no one is really sure why, but a lot of audio stuff is that way), contains an equal amount of energy per octave, which yields a far more useful display. However, the analyzer still displays a summed response of all that it hears, just as a listener would. Analog real-time analyzers (useful tools that have been around for years) can be thought of as 30 or so "tuned" sound-level meters, each displaying a different 1/3-octave band of energy on standardized (a standard that everyone agrees on!) center frequencies. The output is usually displayed on a matrix of LEDs.

A new twist to the RTA game is the FFT real-time analyzer. These little math machines gather data in the time domain (as a function of time) much in the same way you would make a recording. The gathered data is displayed in the frequency domain, a process made possible by a mathematical routine called a Fast Fourier Transform (FFT). Dual-channel FFT analyzers work like single-channel FFT analyzers, with the added advantage of being able to compare the two channels and display the difference between them. The stimulus can be fed to one channel and the output from the measurement microphone into the other. These systems know exactly what was fed into the system and can measure exactly what comes out of the system. As such, the two can be compared and the actual response of the system can be displayed. When this response includes both the time and frequency behavior of the system, it is called a transfer function. Such measurements are invaluable for diagnostic work because they can ignore the room noise by averaging successive measurements. Variable time windows can also be used to allow the direct sound field to be observed without the effects of reflections and reverberation. The stimulus can be any broadband source, including noise and music.

Stimulus-response measurement systems generate a test signal that is fed to the system and reacquired at the output of a system component or at a listener seat with a measurement microphone. Stimuli can include noise-like signals, chirps and swept sine waves, all of which are ways to acquire the impulse response of a sound system without using an impulse. In the simplest sense, an impulse response measurement means "Bang on it and see what happens." Thumping watermelons and kicking tires are crude examples (yes, there's a little scientist in everyone). The simplest form of impulse response is the hand clap. Modern measurement tools are essentially a means of improving on this time-proven method by removing its shortcomings (hand claps aren't perfect or repeatable, and even if they were, a loudspeaker couldn't reproduce them). Our modern generation of analyzers can measure a system's impulse response without using an impulse. It is arrived at indirectly through some mathematical routines. Once the impulse response of a system is known, its response to any other stimuli can be determined.

Thus far, we have only mentioned analyzers that acquire data in the time domain. A notable exception is the TDS analyzer. TDS is an acronym for time delay spectrometry, named by its inventor, Richard Heyser of the Jet Propulsion Laboratories. TDS analyzers gather their data in the frequency domain via a swept sine wave and a synchronized tuned tracking filter. The tracking filter has a narrow bandpass, and synching it with the send signal allows the analyzer to "hear" all of the signal but only a small amount of the noise that is present. The ability to make precision measurements under noisy conditions makes this a valuable tool for many applications.

Remember that all measurement systems require an understanding of audio and acoustic fundamentals to be affective; the radiologist gets the big bucks, not the X-ray technician. After all, what you are really getting paid for is interpreting the data, not just collecting it.

System gain structure Gain structure problems manifest themselves in two ways: one, as audible noise at the system output because of insufficient signal levels through the signal processing chain, or two as audible clipping that results from insufficient headroom in the system. If something sounds "fuzzy" or the meters don't move, you need to check your gain structure.

Establishing a proper system gain structure involves evaluating each electrical device in the system and providing the appropriate impedance and signal level between it and the other components in the system. For each device in the system, it is prudent to know the following information:

Microphones: Output impedance

Sensitivity

Line-level devices: Input impedance

Output impedance

Maximum output voltage

Power amplifiers: Maximum continuous output power

Minimum load impedance

Input sensitivity

Loudspeakers: Sensitivity ratings

Power handling capabilities

With this information in hand, you can bring any system to its full performance potential.

The following is an overview of an orderly approach to establishing the gain structure of the system. It should be performed after the system interconnect procedure described earlier, only this time by starting at the system input and working toward the output.

Verify that each microphone is properly wired and is connected to a load equal to at least 10 times its own output impedance. This intentional impedance mismatch should be maintained at every interface between all system components. It assures that the output voltage of each device remains unchanged when connected to the input of the next device. The microphone should have the proper sensitivity to be able to drive the mixer to a zero indication when picking up the sound source in the manner in which it will be used. This level can be trimmed back if not needed.

In turn, the mixer should be operating in the optimum part of its dynamic range for adequate headroom and optimum signal-to-noise ratio. On most consoles this means that the average output signal from the device is about 1 V (0 dBV) with 20 dB of headroom (clips at 10 VRMS). Actual values will vary from brand to brand (an example of a standard that no one agrees on). As such, a zero indication does not mean the same thing on all meters. I like to measure an unknown mixer's maximum output voltage before clipping as well as its output with a zero meter indication.

Virtually all mixers have a very low output impedance and are connected to very high input impedances, so calibration can be done with open-circuit voltage measurements. To verify that the mixer is operating open-circuit, use the plug in Figure 4 and a voltmeter to verify that the mixer's output voltage does not change when its output is connected to the input of the next device up the line. A significant voltage change indicates excessive loading of the mixer's output usually caused by improper use of Y cords, a pad of improper impedance, shorted wires, etc.

To properly set the mixer's operating level, input a tone to one of the input channels. With the main and channel faders at their zero mark, advance the trim control until the mixer reaches clipping. Oscilloscopes are well suited for this purpose but, just like pens and policemen, it seems like there's never one around when you need one. I prefer to use a small, inexpensive piezoelectric tweeter connected between pins two and three on the output jack to signal the onset of clipping. Such a device has a high impedance and sufficient bandpass to reproduce the harmonics produced by an overdriven device. If a 400 Hz test tone is used, you won't hear anything until the unit clips and the piezo tweeter sounds off. Leave the mixer set just below clip and connect the next device (usually the equalizer). Move the piezo tweeter to the equalizer output and listen for clipping.

Many mixers have much hotter outputs than the input circuits of most equalizers can handle. If the equalizer passes the signal cleanly, advance its gain control until it clips. If the equalizer is clipping with its gain control set at zero (usually the case), insert a pad or attenuator between it and the mixer to reduce the drive level to the equalizer to a value that it can pass without clipping. Proceed in a similar manner until you reach the power amplifier. Crossovers can be calibrated in this way by increasing the bandpass of one of the outputs to pass both the fundamental tone and the harmonics. The gain settings for that channel can be duplicated on the other bandpass outputs. Don't forget to restore the proper bandpass for that output before energizing the loudspeakers.

This procedure assures that each device in the signal chain clips simultaneously and has the same amount of headroom above its average operating level for transients to pass. It also reveals the wide range of internal structures used in our industry (another standard that no one agrees on). In actual use, the mixer will be operated at or near its zero meter indication, and each device after the mixer will have the same headroom that the mixer does. As such, the meter on the mixer displays the operating level of every device in the signal chain.

To adjust the power amplifiers, disconnect the tone generator and input some broadband noise or program material from your favorite CD into the mixer, adjusting the trim control for a zero indication on the main meter. Set the input attenuators on the power amplifiers at minimum and switch on the amplifiers. Advance the input attenuators until the amp clips on the peaks, is loud enough, or your loudspeaker smokes. If the smoke comes first, remember back at the beginning that I said we were proceeding with the assumption of a good design. Everyone knows that when you let the smoke out of a loudspeaker that it never works right anymore.

The amplifier input attenuators will probably be at 12 o'clock or less, indicating that excess drive level is present at the amplifier's input terminals, which is always the case if you have things set right. I realize that this violates one of the 10 commandments of rock and roll, so if you wish to operate the amplifiers wide open, insert a pad or attenuator ahead of the amplifier to reduce the gain overlap between the driving component and the amplifier. Some prefer to simply reduce the drive from the crossover, which in some equipment can reduce the signal-to-noise ratio that we have tried so hard to maintain. But then again, at 110 dB at the back row, who cares?

In short, a properly calibrated signal chain will have some flashing lights. Meters should indicate strong readings and peak lights should flash intermittently. Calibrated in this manner, the system should have a noise floor that is inaudible in the audience area. In fact, you may have to speak into the system to verify that it is even on.

Acoustic gain Sound systems with inadequate acoustic gain will go into feedback before the desired level is produced at the listener. Unfortunately, this important system parameter is best achieved during the design process, where the required acoustic separation between microphones and loudspeakers can be determined and designed into the system.

In auditoriums, a quick way to evaluate whether the acoustic gain of the system is adequate is to use a sound-level meter to measure a live talker at a listener position with and without the use of the sound system. A normal talker produces a sound level LP of 71 dBA at a distance of 2 feet (610 mm). This represents normal face-to-face communication conditions.

The question now is whether the face-to-face level (or higher if necessary) can be achieved at all listening positions when the talker is using the sound system. A good designer brings every listener into a face-to-face relationship with the talker by virtue of the acoustic telescope that we call a sound-reinforcement system. The variable here, once the loudspeakers and microphones are in place, is the distance between talker and microphone, which is why acoustic gain problems usually occur with lectern, floor, lapel and choir microphones.

Several remedies exist for insufficient acoustic gain, including a few that can be worse than the disease. Filters and such can be useful to add some stability to a system but are rarely sufficient for correcting a poor design job. On an installed system, once you've determined for each microphone the maximum talker-to-mic distance that allows adequate acoustic gain, take some time to educate the end-user on how to use the microphone.

Avoiding the worst This article addresses the major parameters that should be evaluated during the installation and testing phase of a project. It is by no means exhaustive, but it will serve to head off many of the major problem areas that seem to plague systems on a regular basis. If your systems are safe, don't hum, have inaudible noise and sufficient acoustic gain, and don't play music when none is desired, your phone should be ringing off the hook with referrals from satisfied clients rather than with threats from disgruntled customers. Adapt a systematic system check-out procedure and stick to it, and you will live long and prosper in the world of pro audio.